of the benefits of digital representation of the analog voice signals, pulse code modulation techniques were integrated into the telephone network. Pulse code modulation (PCM) converts analog sound into digital form by sampling the analog sound so many times per second and converting the sound into a numeric code. After the analog wave form is sampled, it is converted into a discrete digital form, as samples represented by code that indicates the amplitude of the wave form at the instant the sample was taken. A standard telephone form of PCM uses 8 bits for the code and a logarithm compression method that assigns more bits to lower amplitude signals. A standard transmission rate of 64K bits per second is used for one channel of telephone digital communication. The two basic variations of 64K bps PCM are μ-law and A-law. Both methods are similar in that they both use logarithmic compression to achieve 12-13 bits of linear PCM quality with 8 bits. They differ in relatively minor compression details. North America uses μ-law modulation. Europe uses A-law modulation. Another compression method that is often used today is an adaptive differential pulse-code modulation (ADPCM). A commonly used form of ADPCM is ITU-T G.726, which encodes by using 4 bit samples giving a transmission rate of 32K bps. Unlike PCM, the 4 bits do not directly encode the amplitude of speech, but rather the differences in amplitude as well as the rate of change of that amplitude employing rudimentary linear prediction.

Cont
A1

Please replace the second paragraph on page 3 with the following rewritten paragraph.

A2

Another problem experienced in traditional toll networks is echo. Echo is normally caused by mismatch in impedance between the 4-wire network switch conversion to a 2-wire local loop. Although hearing your own voice in the receiver is